REMARKS

By this Amendment, claims 1-5, 7-11, 13-24, and 26-29 have been amended. Thus, claims 1-29 remain pending in the present application.

Claims 1-29 have been rejected under 35 U.S.C. § 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter.

With respect to the Examiner's rejection of claim 1 on the basis that the term "the amplitude" lacks antecedent basis, Applicants note that § 2173.05(e) of the MPEP instructs that "[i]nherent components of elements recited have antecedent basis in the recitation of the components themselves. For example, the limitation 'the outer surface of said sphere' would not require an antecedent recitation that the sphere has an outer surface." In view of such, Applicants respectfully submit that all electrical signals inherently have an amplitude, and hence the claim recitation of "the amplitude" is not indefinite for lack of antecedent basis.

The term "the result" has been amended to the term "the mapped result" to more specifically reference the result of the claimed mapping of the signal from the range of values D onto the range of values W. The term "mapped result" thus refers to the one-to-one translational correspondence between each of the values from the range D into a value within the range W.

With respect to claim 2, again, it is inherent that a range of values will have small values and large values. Also, it is clear that the "emphasis" recited in claim 2 is a further limitation of the "emphasis" recited in claim 1. In other words, while claim 1 recites that "selected value ranges within the first and/or second predetermined range of values" is emphasized, claim 2 more specifically recites that the "selected value ranges" includes the "smallest values of the first range of values."

In claim 9, the phrase "the terms" serves as an introduction to the identified function "a_i*x_{t-i}." One of ordinary skill in the art would readily understand that the phrase "the terms a_i*x_{t-i}"is not intended to reference any previously recited element, and would find that this phrase is clear and definite as recited in the claim. MPEP § 2173.05(e) instructs that

> A claim is indefinite [only] when it contains words or phrases whose meaning is unclear. The lack of clarity could arise where a claim refers to 'said lever' or 'the lever,' where the claim contains no earlier recitation or limitation of a lever and where it would be

unclear as to what element the limitation was making reference. . . . Obviously, however, the failure to provide explicit antecedent basis for terms does not always render a claim indefinite."

Since one of ordinary skill in the art would clearly understand the recitation to the claimed function as presently recited, claim 9 is in fully compliant with the requirements of 35 U.S.C. § 112, second paragraph in this regard. If, however, the Examiner nevertheless continues to object to this language, Applicants respectfully request the Examiner to suggest a suitable remedy for the perceived problem.

Claims 3-5, 7-8, 11, 13-17, and 23-24 have been amended to address the Examiner's concerns.

In view of the foregoing amendments and explanations, Applicants respectfully submit that claims 1-29 clearly and distinctly recite the inventive subject matter, and respectfully requests withdrawal of the rejections under 35 U.S.C. § 112, second paragraph.

Claims 1, 18, 19, 27 and 29 have been rejected under 35 U.S.C. § 102(b) as being anticipated by Kenyon et al., U.S. Patent No. 4,450,531.

Claim 1 is directed to a method for "compressing an electric audio signal" and claims 18-19, 27 and 29 are directed to devices for performing the claimed method. The claimed invention includes the step of "mapping [an] audio signal using a nonlinear function onto a second predetermined range of values W in order to obtain an emphasis of selected value ranges within the first and/or second predetermined range."

Kenyon differs significantly from the claimed invention in that it is not directed to the compression of audio signals and does not map an audio signal from a first predetermined range onto a second predetermined range to obtain an emphasis of a selected value range within the first and/or second predetermined ranges. Instead, Kenyon discloses a method for comparing a broadcast signal to a reference signal. The Examiner points to column 4, lines 36-52 and Fig. 1 as teaching the mapping function recited in Applicants' claims. However, in Kenyon, an additional segment "z" having all zero values is concatenated onto the reference segment D, and then the segment D plus z is Fourier transformed. This is not a mapping of the original range of values as recited in Applicants' claims, because the resulting range of values in Kenyon does not correspond on a one-to-to one basis with the original range of values D. Moreover, in Kenyon, both the reference signals

and the broadcast signals are Fourier transformed, whereupon the two results are then compared in accordance with their respective peak values. Kenyon fails to disclose applying a nonlinear function to map one range of values onto a second range of values to obtain an emphasis of a selected range of values within one of the first and second ranges.

Since Kenyon fails to teach each and every feature of the invention as recited in Applicants' claims, the claimed invention is not anticipated by Kenyon, and withdrawal of this rejection is respectfully requested.

Claims 20- 22 have been rejected under 35 U.S.C. § 103(a) as being unpatentable over Kenyon et al, in view of Uehara, U.S. Patent No. 5,754,798.

Uehara is directed to a power supply controller for a laptop or notebook computer, and is completely unrelated to the broadcast signal comparison method disclosed in Kenyon. Thus, Uehara is insufficient to disclose the features of Applicants' invention found lacking in Kenyon.

Moreover, due to the vastly different nature of the inventions disclosed in Uehara and Kenyon, Uehara is not analogous art with Kenyon. Specifically, there is absolutely no teaching, disclosure, or suggestion in either reference which would compel one of ordinary skill in the art to combine the two references. *See, e.g.*, In re Sponnoble, 405 F.2d 578, 585, 160 USPQ 237, 243 (CCPA 1969) (instructing that the question to be asked is would "the teachings of the prior art . . . in and of themselves and without the benefit of appellant's disclosure, make the invention as a whole obvious").

In view of the foregoing, it is clear that the claimed invention cannot be rendered obvious by the combination of Kenyon and Uehara. Accordingly, withdrawal of this rejection is respectfully requested.

Applicants note that dependent claims 2-17 and 23-26 have not been rejected on their merits. In addition to the features recited in independent claims 1 and 23 which patentably distinguish the invention over the prior art of record, claims 2-17 and 24-26 recite additional features which further distinguish the invention over the prior art. For example, claim 2 recites that the nonlinear function used for mapping the audio signal has a slope dW/dD which decreases with increasing values in the range D to obtain an emphasis of the smallest values of said first range of values. This feature of the invention is nowhere disclosed or suggested in the prior art of record.

As all of claims 1-29 are believed to be allowable over the art of record, Applicants respectfully submit that the present application is currently in condition for allowance, whereupon early and favorable reconsideration in this regard is courteously solicited.

Respectfully submitted,

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APPENDIX A

"CLEAN" VERSION OF EACH PARAGRAPH/SECTION/CLAIM
37 C.F.R. § 1.121(b)(ii) AND (c)(i)

CLAIMS (with indication of amended or new):

A1

1. (Amended) Method for compressing an electric audio signal which is produced in the process of recording ambient noise by means of an electroacoustic transducer, comprising:

normalizing the amplitude of said audio signal or of a digital or analog signal derived therefrom to a first predetermined range D;

mapping said audio signal using a nonlinear function onto a second predetermined range of values W in order to obtain an emphasis of selected value ranges within the first and/or second predetermined ranges; and

storing the mapped result in an electronic memory in a digital format.

SUB C1

2. (Amended) The method of claim 1, wherein said nonlinear function has a slope dW/dD which decreases with increasing values in the range D to obtain an emphasis of the smallest values of said first range of values.

3. (Amended) The method of claim 1, wherein said mapped result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits.

4. (Amended) The method of claim 1, further comprising dividing said audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in said each band signal is present only in an attenuated form.

5. (Amended) The method of claim 4, wherein said audio signal is divided into from 3 to 15 band signals.

7. (Amended) The method of claim 4, wherein the band signals are generated by splitting once or a cascaded multiple of times an input signal which is either the audio signal or an output signal obtained according to the following steps:

A2

Sub F1

- first low pass filtering to generate a first output band signal, and
- subtracting the first output band signal from the input signal to generate a second output band signal.

Sub C3

8. (Amended) The method of claim 7, wherein said low pass filtering is realized by means of a digital convolution over 10-30 values.

9. (Amended) The method of claim 8, wherein for the purpose of the low pass filtering, the convolution is performed using the terms $a_i * x_{t-i}$, wherein the coefficients a_i, $0 < i < 18$, being approximately equal to {0.03, 0.0, -0.05, 0.0, 0.06, 0.0, -0.11, 0.0, 0.32, 0.50, 0.32, 0.0, -0.11, 0.0, 0.06, 0.0, -0.05, 0.0, 0.03}.

10. (Amended) The method of claim 7, wherein the input signal is digitized and only every nth value of each division stage is added to the band signal, n being greater than or equal to 2, in order to compensate for the increased data volume resulting from the splitting into band signals.

Sub B1

11. (Amended) The method of claims 1, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from said audio signal or from a signal derived from said audio signal, said energy signal being generated by squaring said audio signal or said signal derived therefrom.

A3 Sub B2

13. (Amended) The method of claim 12, wherein said second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values, the coefficients of the convolution being essentially equal to each other.

SUB C6

14. (Amended) The method of claim 13, wherein said second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2.

Sub B3

15. (Amended) The method of claim 11, further comprising performing a subsequent differentiation of the energy signal with respect to time to obtain an energy difference signal, said differentiation being performed by computing the difference between two respective values of the energy signal.

16. (Amended) The method of claim 1, wherein the range of normalized values D, is defined by a lower limit D_u D_o, and an upper limit, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u+1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

A3

17. (Amended) The method of claim 1, wherein essentially all steps of the method are performed by integer or fixed point arithmetic.

SUB C8

18. (Amended) Device for carrying out the method of claim 1, comprising a hearing sample unit comprising at least one signal processor for performing at least one processing step of the method.

19. (Amended) The device of claim 18, further comprising a non-volatile semiconductor memory connected to said processor for storing the results of the method.

20. (Amended) The device of claim 18, further comprising a timer connected to a power supply of said hearing sample unit for switching off the hearing sample unit when no processing activity is required.

21. (Amended) The device of claim 19, wherein a power supply of said non-volatile memory and/or said memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the storage of the results in order to reduce the energy consumption by the memory.

22. (Amended) The device of claim 18, wherein the device is an object which is usually carried by persons.

Sub B4 A3

23. (Amended) Method for evaluating recorded hearing samples comprising

recording a plurality of samples of programs to be monitored wherein the samples have at least the same duration as a corresponding plurality of hearing samples,

subjecting the program samples and the hearing samples respectively to the same processing steps, and

calculating a first correlation for comparing the hearing samples with the processed program samples in order to find a match.

24. (Amended) The method of claim 23, wherein the recording of the program samples is started sufficiently before of the hearing samples and the program sample recording is sufficiently longer than that of the hearing samples to ensure that in the correlation, time shifts between the hearing samples and the program samples can be compensated by a displacement in time of the hearing samples with respect to the program samples.

Sub B5 A4

26. (Amended) The method of claim 24, wherein the comparison of the hearing samples with the program samples is effected in two passes, wherein a first pass comprises comparing a respective hearing sample to all program samples using said first correlation the calculation of

which uses a coarse graduation of the time shift, and wherein a second pass comprises using a second, more rugged correlation which provides a finer graduation of the time shift which has a correlation value above a predetermined unit, said second correlation being chosen such that great deviations between the hearing and the program sample have a smaller influence upon the correlation coefficients than in the first correlation, and being effected according to the formula

B5

$$r_t = \frac{\sum_{i=1}^{N} |s_i - a * m_{i-t}|}{\sum_{i=1}^{N} |s_i|}$$

wherein

N : number of hearing sample values used in the correlation,

t : time shift between the hearing and the program sample,

S_i : hearing sample value at the time i,

m_1 : program sample value at the time i, and

a : scaling factor which takes account of the damping of the program signal with respect to the hearing sample;

A4

r_t : correlation value for the shift t, 0 (optimal correlation) < r_t < 1 (no correlation),

a being determined in such a manner that r_t assumes a minimal value.

SUB C1

27. (Amended) A magnetic, optical or magneto-optical data carrier, containing a recorded program which executes the method according to claim 1.

28. (Amended) A magnetic, optical or magneto-optical data carrier, containing a recorded program which executes the method according to claim 23.

Sub E10 A4

29. (Amended) Device comprising at least one program controlled processor unit and a memory for storing a program controlling said processor unit, wherein said memory contains a program which controls at least one of the operations of the method of claim 1.

APPENDIX B

VERSION WITH MARKINGS TO SHOW CHANGES MADE

37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

CLAIMS:

1. (Amended) Method for [the compression of] compressing an electric audio signal which is produced in the process of recording [the] ambient noise by means of an electroacoustic transducer, [more particularly a microphone,] comprising:

[wherein]

normalizing the amplitude of said audio signal or of a [derived] digital or analog signal [is normalized] derived therefrom to a first predetermined range D;

mapping said audio signal [is mapped] using a nonlinear function onto a second predetermined range of values W in order to obtain an emphasis of [sensitive] selected value ranges within the first and/or second predetermined ranges; and

storing the mapped result [is stored] in an electronic memory in a digital format [form].

2. (Amended) The method of claim 1, wherein [a] said nonlinear function [is used whose] has a slope dW/dD which decreases with increasing values in the range D [in order] to obtain an emphasis of the [small] smallest values of said first range of values.

3. (Amended) The method of claim 1, wherein said mapped result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits[, preferably from 4 to 8 bits, and more preferably of 4 bits].

4. (Amended) The method of claim 1, [wherein] further comprising dividing said audio signal [is divided] into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and [each band signal only containing the] wherein any content of the other band signals contained in said each band signal is present only in [a clearly] an attenuated form[, more particularly attenuated to the half, or not at all].

5. (Amended) The method of claim 4, wherein said audio signal is divided into from 3 to 15[, preferably 4 to 10, more preferably 5 to 8, and particularly preferably 6] band signals [are produced].

7. (Amended) The method of claim 4, wherein the band signals are generated by [a single or a cascaded multiple] splitting [of] once or a cascaded multiple of times an input signal which is either the audio signal or [one of the] an output [signals in applying] signal obtained according to the following steps:

- first low pass filtering [generating] to generate a first output band signal, and

- [subtraction of] subtracting the first output band signal from the input signal [for the generation of] to generate a second output band signal[; all first low pass filterings preferably having the same Q-factor].

8. (Amended) The method of claim 7, wherein said low pass filtering is realized by means of a digital convolution over 10-30 values[, preferably 15-25 values, and more preferably 19 values].

9. (Amended) The method of claim 8, wherein for the purpose of the low pass filtering, the convolution is performed [with] using the terms $a_i * x_{t-i}$, wherein the coefficients a_i, $0 \leq i \leq 18$, being approximately equal to {0.03, 0.0, -0.05, 0.0, 0.06, 0.0, -0.11, 0.0, 0.32, 0.50, 0.32, 0.0, -0.11, 0.0, 0.06, 0.0, -0.05, 0.0, 0.03}.

10. (Amended) The method of claim 7, wherein the input signal is digitized and only every nth value of each division stage is added to the band signal, n being [at least] greater than or equal to 2 [and preferably $n = 2$], in order to compensate for the increased data volume resulting from the splitting into band signals.

11. (Amended) The method of claims 1, [wherein] further comprising generating an energy signal which is proportional to [the] an energy content [is generated] of the ambient noise

from said audio signal or from a signal derived [therefrom] from said audio signal, said energy signal [preferably] being generated by squaring said audio signal or said signal derived therefrom.

13. (Amended) The method of claim 12, wherein said second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values, [preferably 40 to 55 values, and more preferably 48 values approximately,] the coefficients of the convolution [preferably] being essentially equal to each other [and more preferably equal to 1.0].

14. (Amended) The method of claim 13, wherein said second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2 [and preferably equal to the number of values of the convolution of the second low pass filtering].

15. (Amended) The method of claim 11, [wherein] further comprising performing a subsequent differentiation of the energy signal with respect to [the] time [is effected in order] to obtain an energy difference signal, said differentiation [preferably] being [effected] performed by computing the difference between [each] two respective values of the energy signal.

16. (Amended) The method of claim 1, wherein the [normalization to a] range of normalized values D, [W, which] is defined by a lower limit [W_u, preferably 0,] D_u D_o, and an upper limit [W_0, where W_0- W_u is preferably equal to 2^n-1, n being a whole number greater than 4 and preferably equal to 7], and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the [input] audio signal or the derived signal within the [normalizing] duration of normalizing the audio or derived signal, which is shorter than or [preferably] equal to the duration of a hearing sample,

- [by] multiplying the reciprocal value of said maximum by [$(W_0$-W_u + 1)] $(D_0$ - D_u+1) , and

-[by] multiplying this product by each value of the [input] audio or derived signal within the duration of the normalized signal.

17. (Amended) The method of claim 1, wherein essentially all steps of the method are performed by integer or fixed point arithmetic[, preferably by binary arithmetic with a number of digits as provided by the employed computing unit].

18. (Amended) Device for carrying out the method of claim 1, [wherein the device includes] comprising a hearing sample unit comprising at least one signal processor [which memory is destined to perform] for performing at least one processing step of the method.

19. (Amended) The device of claim 18, [wherein] further comprising a non-volatile semiconductor memory [is] connected to said processor [which allows to store] for storing the results of the method.

20. (Amended) The device of claim 18, [wherein] further comprising a timer [is] connected to [the] a power supply of said hearing sample unit [which allows to switch] for switching off the hearing sample unit when no processing activity is required, [more particularly in the periods between the processing of two hearing samples, in order to reduce the energy consumption].

21. (Amended) The device of claim [20] 19, wherein [the] a power supply of said non-volatile memory and/or said memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the storage of the results in order to reduce the energy consumption by the memory.

22. (Amended) The device of claim 18, wherein [it is in the form of] the device is an object which is usually carried by persons[, preferably in the form of a wristwatch].

23. (Amended) Method for [the evaluation of the results of the] evaluating recorded hearing [sample] samples [processing according to claim 1, wherein program] comprising

recording a plurality of samples of [the monitored] programs [are recorded which] to be

<u>monitored wherein the samples</u> have at least the same duration as [the] <u>a corresponding plurality of</u> hearing samples,

<u>subjecting</u> the program samples [are subjected] <u>and the hearing samples respectively</u> to the same processing steps [as the hearing samples], and [a calculation of]

<u>calculating</u> a first correlation [of] <u>for comparing</u> the hearing samples with the processed program samples [is effected] in order to find a match.

24. (Amended) The method of claim 23, wherein the recording of the program samples is started sufficiently before [that] of the hearing samples and [its] <u>the program sample recording</u> is sufficiently longer than that of the hearing samples to ensure that in the correlation, time shifts between the [timer for the] hearing samples and [the timer for] the program samples can be compensated by a displacement in time of the hearing samples with respect to the program samples.

26. (Amended) The method of claim 24, wherein the comparison of the hearing samples with the program samples is effected in two passes, <u>wherein a first pass comprises comparing</u> a respective hearing sample [being compared] to all program samples [in all ways in the first pass by means of] <u>using</u> said first correlation [whose] <u>the</u> calculation <u>of which uses</u> [is simpler due to] a [coarser] <u>coarse</u> graduation of the time shift, <u>and wherein a second pass comprises using</u> [while in the case of a time shift whose correlation values c_t are above a predetermined limit,] a second, <u>more</u> rugged correlation [is effected] which provides a finer graduation of the time shift <u>which has a correlation value above a predetermined unit</u> [and in particular, a time resolution which is at least twice as high as in the first correlation], said second correlation [preferably] being chosen such that great deviations between the hearing and the program sample have a smaller influence upon the correlation coefficients than in the first correlation, and [preferably] being effected according to the formula

$$r_t = \frac{\sum_{i=1}^{N} |s_i - a * m_{i-t}|}{\sum_{i=1}^{N} |s_i|}$$

[where] <u>wherein</u>

N : number of hearing sample values used in the correlation,

t : time shift between the hearing and the program sample,

S_i : hearing sample value at the time i,

m_i : program sample value at the time i, and

a : scaling factor which takes account of the damping of the program signal with respect to the hearing sample;

r_t : correlation value for the shift t, 0 (optimal correlation) $\leq r_t \leq$ 1 (no correlation),

a being determined in such a manner that r_t assumes a minimal value.

27. (Amended) [Data carrier, more particularly] <u>A</u> magnetic, optical or magneto-optical data carrier, containing a recorded program [upon whose execution] <u>which executes</u> the method according to claim 1 [is carried out].

28. (Amended) [Data carrier, more particularly] <u>A</u> magnetic, optical or magneto-optical data carrier, containing a recorded program [upon whose execution] <u>which executes</u> the method according to claim 23 [is carried out].

29. (Amended) Device comprising at least one program controlled processor unit and a memory for [the storage of the] <u>storing a</u> program controlling said processor unit, wherein said memory contains a program [under whose control] <u>which controls</u> at least one [and preferably all] <u>of the</u> operations of the method of claim 1 [can be performed].